**ECE 311**
**Fall of 2017**
**Final Lab**
**12/12/2017 8 AM CT**
**Time Limit: 72 Hours**

**Name:** ______________________________

Section (A / B/ C)

- This final lab contains 6 pages (including this cover page) and 5 questions. Total of points is 100.
- This lab must be submitted into **Compass 2g** by **8 AM Friday December 15, 2017 Central Time**.
- There will be **no extensions** for this exam.

Grade Table (for TA use only)

| Question | Points | Score |
|---|---|---|
| 1 | 20 | |
| 2 | 20 | |
| 3 | 20 | |
| 4 | 20 | |
| 5 | 20 | |
| Total: | 100 | |

1. **Spectral Analysis**. In this problem you'll perform analysis on a song sample. The data for this problem is in **song1.mat** in variable sigp. This is a song that has been contaminated by a very strong single pitch tone.

   (a) (10 points) Load **song1.mat**. A variable sigp should appear in your workspace. This song was sampled at 44.1 KHz. Plot the magnitude and phase spectrum of this song in a two panel plot. Your frequency axis should be in Hz units.

   (b) (5 points) Identify the frequency of the contaminating pitch. Explain how this was determined.

   (c) (5 points) Would a low pass filter be reasonable to use to remove the pitch? Explain. Write down any underlying assumptions.

2. **Filter Design** Ideal filter, truncated sinc filter and designed filter using PM method.

   (a) (5 points) Recall that an ideal high-pass filter will have the shape like in Figure. 1.

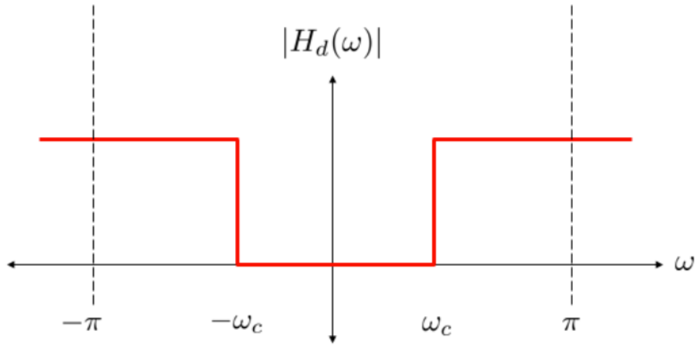


Figure 1: An ideal high-pass filter.

$$h[n] = \delta[n] - \frac{\omega_c}{\pi} sinc(\frac{\omega_c}{\pi} n) \qquad (High - Pass) \tag{1}$$

Let $n = -N : N$, where $N = 100$. Also, in this case, we want to have a cutoff frequency $\omega_c = \frac{2\pi}{3}$. Use the equation provided above, plot the time-domain impulse response using **stem** and the frequency-domain magnitude response (in dB) using **plot**

   (b) (15 points) Given a sample rate fs = 18,000 Hz, design a **minimum-length, linear phase high-pass filter** with passband edge at 6 kHz and stopband edge at 5.6 kHz. The passband ripple should be no more than 1 dB and the stopband attenuation should be less than 50 dB. Plot the magnitude and phase response of the filter and add markers in your plot to prove that it meets the required specifications. What is the length of your filter?

3. **Image Processing** As a DSP Engineer you are to determine the best image filter for two applications. You will apply these filters on two images : grumpycat.jpg and coins.png.

   (a) (5 points) To hide the identity of the cat in 'grumpycat.jpg', you would like to blur the image. Would you use a LPF or HPF? Explain.

   (b) (5 points) Implement the filter you identified in part a on 'grumpycat.jpg'. You can read in this image with imread. You need only apply the filter on the first channel of the image. As output, display the filtered image, next to the original image (two panel plot).

   (c) (5 points) To find the position of coins for a robotics application, you would like to detect the edges in a coin image. Would you use a LPF or HPF? Explain.

   (d) (5 points) Implement the filter you identified in part c on 'coins.png'. You can read in this image with imread. As output, display the filtered image, next to the original image (two panel plot).

4. **Multi-rate DSP** Given a portion of a sound track, you are asked to demonstrate downsampling and upsampling during an interview. **LABEL AXES IN Hz**

   (a) (5 points) Load the soundtrack named **Radiohead_chopped.wav**. Listen to it and plot out its frequency magnitude spectrum.(in normal scale). The sampling frequency for this song is 48 kHz.

   (b) (5 points) Design a Low-pass filter that has a cutoff frequency at $\omega_c = \frac{\pi}{3}$ rad/sample. Plot out its frequency response, both magnitude (in dB) and phase.

   (c) (5 points) Now using the LPF designed in the previous part, downsample the **original signal** by 3. Plot the frequency magnitude response for the signal **after filtering** and the frequency magnitude response for the signal **after downsampling**.

   (d) (5 points) Now using the same LPF, upsample the **original signal** by 3. Plot the frequency magnitude response for the signal **before filtering** and the frequency magnitude response for the signal **after filtering**.

5. **Spectrograms and Chirps** . An Audio engineer at Cirrus Logic asks you to analyze the following waveform with a spectrogram. The waveform was generated from samples of $sin(\frac{1}{2} \times \Omega_D t^2)$. Note the instantaneous frequency of this wave form is $\Omega_{inst} = \Omega_D t$. This waveform was sampled at 8192 Hz. $\Omega_D$ is 4000 $\frac{rad}{s^2}$. Be careful of units.

   (a) (5 points) Determine the time when this signal begins to alias. Please explain the process of determining this value. Listen to the signal and describe it.

   (b) (10 points) This waveform is stored in waveform1.mat in a variable called sig. Make a spectrogram of the waveform with following parameters: rectangular windows of length 500, noverlap of 250, and FFT size of 512. You may make use of any functions to do this. Please display this as an imagesc plot and label the axes.

   (c) (5 points) Describe what you see in the spectrogram. Why are there zig-zags?